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EXAMINER	
KHOO, FOONG LIN	

ART UNIT	PAPER NUMBER
2664	

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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)	
	10/010,682	CHU ET AL.	
	Examiner	Art Unit	
	F. Lin Khoo	2664	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 08 November 2001.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☐ Claim(s) _____ is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-48 (previously claims 1-46) is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Specification

1. The disclosure is objected to because of the following informalities: In page 13, line 30 (last line on page 13), SIP server is labeled as number 10 but in Fig. 9 the SIP server is labeled as number 11.

Appropriate correction is required.

Claim Objections

2. The numbering of claims is not in accordance with 37 CFR 1.126 which requires the original numbering of the claims to be preserved throughout the prosecution. When claims are canceled, the remaining claims must not be renumbered. When new claims are presented, they must be numbered consecutively beginning with the number next following the highest numbered claims previously presented (whether entered or not).

Claims 1-46 have been renumbered as 1-48, respectively. Misnumbered claim 2.5, 14.5 have been renumbered to 3 and 16, respectively.

Claim Rejections - 35 USC § 102

3. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

4. Claims 1-14 (previously claims 1-13) are rejected under 35 U.S.C. 102(e) as being anticipated by Vo et al. (U.S. Patent No. 6,795,444).

Regarding Claim 1, Vo et al. discloses a system and method (a software radio port device) for processing Voice over IP (VoIP) data packets for wireless terminals (col 2, lines 58-66), (the device) comprising: an air interface (Fig. 1 elements 110, 114A through 114C, col 11, lines 11-15); an IP/Ethernet Interface (Fig. 2A element 118B, col 12, lines 21-30 and col 9, lines 55-57. IWF is an Interworking Function that provides an interface to manage call control, mobility and other services in relation to cellular transport over IP); a VoIP Media Gateway interposed between the air interface and the IP/Ethernet Interface for media conversion and transportation (Fig. 3A, element 19, col 18, lines 9-13); a VoIP signaling Gateway for controlling VoIP call processing (Fig. 2B and Fig. 2C, col 15, lines 23-40); and, a Call Control for controlling call processing of wireless terminals and coordinating with VoIP call processing (Fig. 2A, element 206, col 14, lines 44-47).

Regarding Claim 2, Vo et al. discloses wherein the air interface receives signaling messages and a voice stream from a mobile station (Fig. 2A, elements 204,

114A and 112A, col 11, lines 65-67 and col 12, lines 1-7. Vo et al. discloses an ANSI-136 RAN (Radio Access Network) air interface to receive signaling and voice stream from a mobile station and it is inherent that a two-way communication path is set up via the air interface and mobile station to exchange voice and signaling information between the two).

Regarding Claim 3 (previously Claim 2.5), Vo et al. discloses wherein the mobile station receives signaling messages and a voice stream from the air interface station (Fig. 2A, elements 114A and 112A, col 11, lines 65-67 and col 12, lines 1-7. Vo et al. discloses an ANSI-136 RAN (Radio Access Network) air interface to transmit signaling and voice stream to a mobile station and it is inherent that a two-way communication path is set up via the air interface and mobile station to exchange voice and signaling information between the two).

Regarding Claim 4 (previously Claim 3), Vo et al. discloses wherein the Call Control (Fig. 2A, element 206, col 14, lines 44-47) is configured for: receiving signaling messages from the air interface (Fig. 2A, element 298, col 12, lines 1-7); instructing the VoIP Media Gateway to set up RTP paths to the called parties (Fig. 11 elements 1506 and 1508, col 25, lines 52-56. Refer to Call Setup message between GW-11 and GW-2. Vo et al. discloses a wireless Internet Protocol (WLIP) network system, having a Circuit Switched Network (CSN) wireless portion and a Packet-Switched Network (PSN) VoIP portion (col 9, lines 1-3). Vo et al. does not disclose a Real Time Transport Protocol

(RTP). It is inherent that cellular wireless networks supports VoIP telephony according to an industry standard, e.g., H.323, SIP, RTP, etc. as disclosed by Barany et al. (U.S. Pub. No. 2001/0043577) in paragraph [0095], lines 5-7 and paragraph [0148], lines 1-10); and instructing the VoIP Signaling Gateway to set up VoIP calls to the called parties (Fig. 11, elements 1502,1504,1510,1512,1514,1516,1518,1520,1522, col 25, lines 52-65).

Regarding Claim 5 (previously Claim 4), Vo et al. discloses wherein the VoIP Signaling Gateway is configured for receiving messages from the Call Control; processing messages from the Call Control; and, managing VoIP call-related activities (Fig. 2A, element 299B, 206, 118B, 221 and 122 , col 12, lines 18-21, lines 28-30. The Call Control (206) is connected to the IWF (118B) which receives and processes messages from the Call Control. The GW-VLR entity 221 has the functions of the Mobility Gateway (Signaling Gateway, see Fig. 2B elements 13 and 120) which handles signaling information between the circuit switched network and packet switched network and has a Visitor Location Register (VLR) for maintaining visiting mobile terminal location information. The VoIP entity 122 handles the VoIP traffic (signaling+user data (voice or otherwise)) with one or more associated servers provided in the PSN infrastructures).

Regarding Claim 6 (previously Claim 5), Vo et al. discloses wherein the VoIP Media Gateway is configured for: receiving messages from the Call Control

processing messages from the Call Control (Fig. 2C, col 15 lines 39-41, Fig. 2A, elements 299B, 206, 118B, col 12, lines 18-21, lines 23-25. The Call Control (206) is connected to the IWF (118B) which receives and processes messages from the Call Control. Fig. 2C shows a gateway 19 which includes the media gateway functionality and signaling gateway functionality integrated together. Therefore, in Fig. 2A the Media GW (116) can be integrated together with GW-VLR (221) and act as one unit within IWF (118B)); receiving the voice stream from the air interface (Fig. 1, elements 114A, 110 and 140, col 11, lines 18-20); and packetizing the voice stream into RTP data packets (col 3, lines 65-67 and col 4, line 1. A media gateway is disposed between the cellular network portion and the packet-switched network portion for providing a communication path between the two and it would be inherent that since the media gateway is connected to a packet switched network, it would therefore perform the function of packetizing the voice stream from the cellular network for distribution into the packet switched network. Vo et al. discloses a wireless Internet Protocol (WLIP) network system, having a Circuit Switched Network (CSN) wireless portion and a Packet-Switched Network (PSN) VoIP portion (col 9, lines 1-3). Vo et al. does not disclose a Real Time Transport Protocol (RTP). It is inherent that cellular wireless networks supports VoIP telephony according to an industry standard, e.g., H.323, SIP, RTP, etc. as disclosed by Barany et al. (U.S. Pub. No. 2001/0043577) in paragraph [0095], lines 5-7 and paragraph [0148], lines 1-10).

Regarding Claim 7 (previously Claim 6), Vo et al. discloses wherein the IP/Ethernet Interface receives RTP data packets from the VoIP Media Gateway and messages from the Call Control and VoIP Signaling Gateway, and sends the RTP data packets and the messages to the packet data network (Fig. 2A, elements 118B, 206, col 12, lines 23-25 col 14, lines 43-46. Fig. 3A, elements 19, 108, col 18 line 2, col 18 line 12. In Fig. 2A IWF (118B) is the IP interface that receives data packets from the VoIP Media Gateway (Fig 3A, element 19) and messages from the Call Control (Fig. 2A, element 206) and VoIP Signaling Gateway (Fig. 3A, element 19) and sends the RTP data packets and messages to the packet data network (Fig. 3A, element 108). Vo et al. discloses a wireless Internet Protocol (WLIP) network system, having a Circuit Switched Network (CSN) wireless portion and a Packet-Switched Network (PSN) VoIP portion (col 9, lines 1-3). Vo et al. does not disclose a Real Time Transport Protocol (RTP). It is inherent that cellular wireless networks supports VoIP telephony according to an industry standard, e.g., H.323, SIP, RTP, etc. as disclosed by Barany et al. (U.S. Pub. No. 2001/0043577) in paragraph [0095], lines 5-7 and paragraph [0148], lines 1-10).

Regarding Claim 8 (previously Claim 7), Vo et al. discloses wherein the IP/Ethernet Interface receives messages and RTP packets from a packet data network (Fig. 2A elements 138, 118B, col 11, lines 36-38) sends the RTP packets to the VoIP Media Gateway (Fig. 2A elements 138, 221 (see Fig 2C where media gateway(11) and signaling gateway(13) are integrated together to form element 221 in IWF(118B) in Fig.

2A), col 11, lines 36-39), and sends the messages to the Call Control and VoIP Signaling Gateway (Fig. 2A, IWF (118B) sends messages to the connected element 206 (CCF)) and VoIP Signaling Gateway (Fig. 2A elements 138, 221 (see Fig. 2C where media gateway(11) and signaling gateway(13) are integrated together to form element 221 in Fig. 2A). Note: RTP is inherent as disclosed in Claims 4, 6 and 7)

Regarding Claim 9 (previously Claim 8), Vo et al. discloses wherein the Call Control is configured for: receiving signaling messages from the IP/Ethernet interface and the VoIP Signaling Gateway; and managing mobile station-related activities (Fig. 2C, col 15 lines 39-41, Fig. 2A, elements 299B, 206, 118B, col 12, lines 18-21, lines 23-25, col 14, lines 24-29. The Call Control (206) is connected to the IWF (118B) which receives and processes messages from the Call Control. Fig. 2C shows a gateway 19 which includes the media gateway functionality and signaling gateway functionality integrated together. Therefore, in Fig. 2A the Media GW (116) can be integrated together with GW-VLR (221) and act as one unit within IWF (118B)).

Regarding Claim 10 (previously Claim 9), Vo et al. discloses wherein the VoIP Signaling Gateway is configured for: receiving messages from the IP/Ethernet interface; instructing the Call Control to manage mobile calls; and managing VoIP call-related activities (Fig. 2A, elements 299B, 206, 118B, 221 and 122 , col 12, lines 18-21, lines 28-30, col 14, lines 24-29. The Call Control (206) handles mobile calls is connected to the IWF (118B) which has the VoIP Signaling Gateway. The GW-VLR entity 221 has

the functions of the Mobility Gateway (Signaling Gateway, see Fig. 2B elements 13 and 120) in the IWF(118B) which handles signaling information between the circuit switched network and packet switched network and has a Visitor Location Register (VLR) for maintaining visiting mobile terminal location information. The VoIP entity 122 handles the VoIP traffic (signaling+user data (voice or otherwise)) with one or more associated servers provided in the PSN infrastructures).

Regarding Claim 11 (previously Claim 10), Vo et al. discloses wherein the VoIP Media Gateway is configured for: receiving the VoIP data packets from the IP/Ethernet Interface; and converting the VoIP data packets to a voice stream (Fig. 2A, elements 108, 118B, 138, 221, 298, 204, col 12, lines 18-31. The VoIP data packets is received from VoIP network (108), connected via path 138 to IWF (118B) where it is converted into voice stream (example shown in Fig. 1 element 140, col 11, lines 18-20) for transfer to RNC (204) coupled to ANSI-136 path 298).

Regarding Claim 12 (previously Claim 11), Vo et al. discloses wherein the Air Interface receives a voice stream from the VoIP Media Gateway and receives signaling messages from the Call Control (Fig. 2A elements 118B, 221, 206, 298, 204, col 12, lines 1-5, col 14, lines 23-47)

Regarding Claim 13 (previously Claim 12), Vo et al. discloses a wireless telecommunication system for providing VoIP service to wireless terminals comprising:

a software radio port device according to claim 1 (col 5, lines 53-56, col 22, lines 22-30); a network server platform (Fig. 6 elements 1006 (gatekeeper, col 22, lines 50-55), 298 (Home Location Register (HLR), 1002 (Mobile Switching Application (MSA), col 22, lines 25-30); a VoIP-enabled communication device (Fig. 6 element 808, col 22, line 38-41); a VoIP proxy server for managing requests/messages from the VoIP-enabled communication device (col 10, lines 65-67, col 11, lines 1-11); and a PSTN/VoIP Gateway for interconnecting a VoIP network with Public Switched Telephone Network (PSTN) (Fig. 2A, element 116, 108, 102, col 10, lines 23-26).

Regarding Claim 14 (previously Claim 13), Vo et al. discloses the network server platform (provides call control, mobility management, subscriber services to the software defined radio base station (SDR BS) as shown in Fig. 6 (col 5, lines 53-56) comprises: an IP interface; a Mobile Switching Center/Visitor's Location Register (MSC/VLR) configured to provide call control operations for mobiles (col 22, lines 25-30 refers to a Mobile Switching Application (MSA). The MSA is IP compatible, therefore it is inherent that it has an IP interface. MSA is essentially the switching control part of a legacy MSC, therefore it is inherent that it is associated with a Visitor Location Register (VLR) (col 4, lines 21-24). The MSA provides all call control functionality); a Home Location Register (HLR) for storing mobile subscriber authentication data (Fig. 6, element 298 (Home Location Register (HLR), col 5 lines 1-18); and a VoIP Call-Server Control to provide VoIP call processing control operations (Fig. 6, element 1006, col 22, lines 50-55)).

5. Claims 25-33 (previously claims 23-31), 45 (previously claim 43), 46 (previously claim 44), 47 (previously claim 45), 48 (previously claim 46) are rejected under 35 U.S.C. 102(e) as being anticipated by Sayers et al. (U.S. Patent No. 6,539,237).

Regarding Claim 25 (previously Claim 23), Sayers et al. discloses a method of providing a two-way voice path between a VoIP device in a network and a mobile station wherein a call originates at the VoIP device, the method comprising: processing a call connection request at a VoIP call-server (Fig. 12, col 21, lines 63-67 and col 22, lines 1-3. In Fig. 12 the H.323-SETUP[H.245.TSAP,BC,CPN,CPG,CRV], SETUP[CPN, CPG], Request a Page for the Called MS are a call connection request at a VoIP call-server. Note: H.323 relates to VoIP protocol, therefore the Gatekeeper is the VoIP call-server); initiating mobile call set-up at a Network Server Platform (NSP) (Fig. 12, col 22, lines 4-21. Phase 2 and Phase 3 discuss the process of initiating a mobile call set-up at a NSP starting with the Paging Request and ending when the Call Confirmed message is received from the MS. Note: Fig. 6 elements 6-1 through 6-15, 6-17, 6-25 (col 14, lines 50-67 and col 15, lines 1-48) and 6-26, together performs the function of the NSP.); tuning the mobile station to a digital traffic channel (DTC) to establish a voice path over the air via a Software Radio Port (SRP) (Fig. 12, col 22, lines 22-27. Phase 4 discusses the Terminal Capabilities, Master/Slave processes and assignment of the mobile station to the traffic channel (TCH) shown in Fig. 12 by Asg CMD and Asg COM (refer to Fig. 10 Asg CMD and Asg COM). This is associated with

tuning the mobile station to digital traffic channel to establish a voice path over the air via the SRP. Note: SRP consists of the NSP elements and includes the air interface modules as defined in Fig. 3 and Fig. 5 for wireless communication with the mobile station); alerting both the mobile station and the VoIP device (Fig. 12 col 22, lines 28-33. Phase 5 discusses the mobile Alerting the originating endpoint (VoIP device) and the Asg CMD can be associated with Alerting the MS); establishing an RTP media path for exchange of RTP data packets via the SRP (Fig. 12, col 22, lines 22-27. Phase 4 discusses B-Open Chn with RTP TSAP which provides RTP TSAP address to be used for actual data transfer); and interconnecting the voice path over the air and the RTP path over the packet network via the SRP (Fig. 13 shows RTP/RCTP Channel open between Endpoints for communication via the SRP).

Regarding Claim 26 (previously Claim 24), Sayers et al. discloses wherein the step of processing a call connection request comprises receiving a call connection request message from the VoIP device and engaging an NSP to analyze the called number (Fig. 12, col 21, lines 63-65, col 22, lines 1-21. In Fig. 12 the H.323-SETUP[H.245.TSAP,BC,CPN,CPG,CRV], SETUP[CPN, CPG], Request a Page for the Called MS are a call connection request at a VoIP call-server. Note: H.323 relates to VoIP protocol, therefore the Gatekeeper is the VoIP call-server. Phase 2 and Phase 3 discuss the process of initiating a mobile call set-up at a NSP starting with the Paging Request and ending when the Call Confirmed message is received from the MS. Note: Fig. 6 elements 6-1 through 6-15, 6-17, 6-25 (col 14, lines 50-67 and col 15, lines 1-48)

and 6-26 together performs the function of the NSP. Engaging the NSP to analyze the called number is associated with the following: (1) the CPN number in the SETUP message is used to lookup up paging information for the Private mobile station (2) authentication and ciphering procedures are performed (3) the RAS layer perform the ARQ sequence with the registered gatekeeper in order to reserve the required bandwidth (4) the mobile station receiving a SETUP message including the Bearer Capabilities for the call being established to make sure the mobile station can correctly determine the type of call).

Regarding Claim 27 (previously Claim 25), Sayers et al. discloses wherein said step of initiating mobile call set-up comprises: verifying the called party as a valid mobile station (Fig. 12, Col 22, lines 4-11. Phase 2 discusses authentication procedure to verify the called party as a valid mobile station); sending a message to page the mobile station via the SRP (Fig. 12, Col 22, lines 4-11. In Phase 2, a Paging Request is sent to the MS via the SRP. Note: SRP consists of the NSP elements and includes the air interface modules as defined in Fig. 3 and Fig. 5 for wireless communication with the mobile station); receiving page response from the mobile station (Fig. 12, Col 22, lines 4-11. As part of Phase 2, a paging response from the MS is associated with a paging request (see message Paging Response[IMSI,Cksn] in Fig. 12)); and instructing the VoIP call-server to forward the call connection request to the SRP (Fig. 12, Col 22, lines 4-11. In Phase 2, the ARQ and ACF instruct the VoIP call-server to forward the

call connection request to the SRP. Note: H.323 relates to VoIP protocol, therefore the Gatekeeper is the VoIP call-server).

Regarding Claim 28 (previously Claim 26), Sayers et al. discloses wherein said step of tuning comprises: sending a message to tune the mobile station to a specified digital traffic channel (Fig. 12, col 22, lines 22-27. Phase 4 discusses the Terminal Capabilities, Master/Slave processes and assignment of the mobile station to the traffic channel (TCH) shown in Fig. 12 by Asg CMD and Asg COM (refer to Fig. 10 Asg CMD and Asg COM). This is associated with tuning the mobile station to digital traffic channel); and detecting the mobile station as being tuned to the specified digital traffic channel (Fig. 12, col 22, lines 28-33. In Phase 5, the ALERTING, H.323-Alerting, CONNECT, H.323-Connect and CONNECT ACK messages are the processes involved in detecting the mobile station as being tuned to the specified digital traffic channel).

Regarding Claim 29 (previously Claim 27), Sayers et al. discloses wherein said step of alerting comprises: sending a message to the mobile station for alerting a mobile user (In Fig. 12 the Asg CMD can be associated with Alerting the MS); and sending a ringing indication to the VoIP device via the VoIP call-server (Fig. 12 col 22, lines 28-33. The H.323-Alerting message can be associated with sending a ringing indication to the EP (VoIP device) via VoIP call-server).

Regarding Claim 30 (previously Claim 28), Sayers et al. discloses wherein said step of establishing comprises: receiving a connect indication from the mobile station; sending the connect indication to the VoIP device via the VoIP call-server (Fig. 12 col 22, lines 28-33. The CONNECT and H.323-Connect shows the connect indication from the MS to the EP (VoIP device) via the VoIP call-server); setting up an RTP media path for exchange of RTP data packets (Fig. 12, col 22, lines 22-27. Phase 4 discusses B-Open Chn with RTP TSAP which provides RTP TSAP address to be used for actual data transfer); and informing the call connection to the NSP (In Fig. 12, the CONNECT to the CM indicates the call connection where the CM is part of the NSP).

Regarding Claim 31 (previously Claim 29), Sayers et al. discloses wherein said step of interconnecting comprises: converting received voice frames to RTP packets to be sent to the packet network, and converting received RTP packets to voice frames to be sent to the mobile station (Fig. 3 - Col 10, lines 56-59, Fig. 5 - Col 11, lines 55-60 and Fig. 10 and Fig. 12 – Col 22, lines 22-27. Air Interface (31-1) /GSM Um-Interface (31-1') and Network Interface (31-2)/H.323 Terminal Interface (31-2') converts voice frames into packets and converts received packets to voice frames for transmission in the RTP/RTCP transport link between the endpoints as shown in Fig. 13 (MS and EP)).

Regarding Claim 32 (previously Claim 30), Sayers et al. discloses wherein the network is a Public Switched Telephone Network (PSTN) (Fig. 4, element 26 (PSTN), col 11, lines 37-43).

Regarding Claim 33 (previously Claim 31), Sayers et al. discloses wherein the network is a Private Branch Exchange (PBX) (Fig. 4, element 43 (PBX), col 11, lines 37-43).

Regarding Claim 45 (previously Claim 43), Sayers et al. discloses a method for terminating a call between a mobile station and a VoIP device in a network comprising: receiving a release indication from the mobile station (Fig. 11 - DISCONNECT from MS indicates receiving a release indication from the mobile station); releasing radio resources and an RTP media path (Fig.11 - Release Channel, EndSession, Channel Released, RELEASE, Channel Release and RELEASE COMPLETE indicate releasing radio resources and an RTP media path); sending a call release request to the VoIP device via a VoIP call-server (Fig. 11 – Disconnect, Release Channel, EndSession indicate sending a call release request to the VoIP device (EP) via a VoIP call-server (gatekeeper)); and sending a call release indication to a Network Server Platform (NSP) (Fig. 11 – EndSession, Channel Released, DCF[Q931.TSAP,32Kkps] in response to a DRQ[IMSI,CPN,32Kkps,CRV] indicate sending a call release indication to a Network Server Platform (NSP). Note: Fig. 6 elements 6-1 through 6-15, 6-17, 6-25 (col 14, lines 50-67 and col 15, lines 1-48) and 6-26, together performs the function of the NSP).

Regarding Claim 46 (previously Claim 44), Sayers et al. discloses a method for terminating a call between a mobile station and a VoIP device in a network comprising:

receiving a release indication from the VoIP device via a VoIP call-server (Fig. 13 – EndSession from EP (VoIP device) indicates receiving a release indication from the VoIP device via a VoIP call-server (gatekeeper)); sending a call release request to the mobile station (Fig. 13 – Release Channel, Disconnect and DISCONNECT indicate sending a call release request to the mobile station); releasing radio resources and an RTP media path (Fig. 13 – RELEASE, Release Resources, DCF[Q931.TSAP,32Kkps] in response to a DRQ[IMSI,CPN,32Kkps,CRV], RELEASE COMPLETE indicate releasing radio resources and an RTP media path); and sending a call release indication to a Network Server Platform (NSP) (Fig. 13 – EndSession, DCF[Q931.TSAP,32Kkps] in response to a DRQ[IMSI,CPN,32Kkps,CRV], and H.323-Release Complete[CRV] indicate sending a call release indication to a Network Server Platform (NSP). Note: Fig. 6 elements 6-1 through 6-15, 6-17, 6-25 (col 14, lines 50-67 and col 15, lines 1-48) and 6-26, together performs the function of the NSP).

Regarding Claim 47 (previously Claim 45), Sayers et al. discloses a method for terminating a call between a first mobile station and a second mobile station **(Note: first mobile station (origination) to second mobile station (termination) can be explained using the combination of Fig. 10 and Fig. 11 (relates to the first mobile station) and Fig. 12 and Fig 13 (relates to the second mobile station))**, said first mobile station associated with a first Software Radio Port (SRP) **(Note: SRP consists of the NSP elements and includes the air interface modules as defined in Fig. 3 and Fig. 5 for wireless communication with the mobile station. Note: Fig. 6**

elements 6-1 through 6-15, 6-17, 6-25 (col 14, lines 50-67 and col 15, lines 1-48) and 6-26, together performs the function of the NSP) and said second mobile station associated with a second SRP, the method comprising:

receiving a release indication at the first SRP from the first mobile station (Fig. 11 - DISCONNECT from MS indicates receiving a release indication at the first SRP from the first mobile station);

releasing radio resources and an RTP media path at the first SRP ((Fig.11 - Release Channel, EndSession, Channel Released, RELEASE, Channel Release and RELEASE COMPLETE indicate releasing radio resources and an RTP media path at the first SRP);

sending a call release request from the first SRP to a VoIP call-server (Fig. 11 – DISCONNECT, Disconnect, Release Channel and DRQ[IMSI,CPN,32Kkps,CRV] indicate sending a call release request from the first SRP to a VoIP call-server (gatekeeper));

sending a call release indication from the first SRP to a Network Server Platform (NSP) (Fig. 11 – Release Channel, EndSession, DCF[Q931.TSAP,32Kkps] in response to a DRQ[IMSI,CPN,32Kkps,CRV] indicate sending a call release indication from the first SRP to a Network Server Platform (NSP));

receiving a release indication at the second SRP from the VoIP call-server (Fig. 13 – EndSession, Release Channel indicate receiving a release indication at the second SRP from the VoIP call-server);

sending a call release request from the second SRP to the second mobile station (Fig. 13 – Disconnect, DISCONNECT indicate sending a call release request from the second SRP to the second mobile station);

releasing radio resources and an RTP media path at the second SRP (Fig. 13 – RELEASE, Release Resources, DCF[Q931.TSAP,32Kkps] in response to a DRQ[IMSI,CPN,32Kkps,CRV], RELEASE COMPLETE indicate releasing radio resources and an RTP media path at the second SRP); and

sending a call release indication from the second SRP to NSP (Fig. 13 – RELEASE, Release Resources, DCF[Q931.TSAP,32Kkps] in response to a DRQ[IMSI,CPN,32Kkps,CRV], H.323-Release Complete[CRV] indicate sending a call release indication from the second SRP to NSP).

Regarding Claim 48 (previously Claim 46), Sayers et al. discloses a method for maintaining an RTP media path during handoff of a mobile station from a first Software Radio Port (SRP) **(Note: SRP consists of the NSP elements and includes the air interface modules as defined in Fig. 3 and Fig. 5 for wireless communication with the mobile station. Note: Fig. 6 elements 6-1 through 6-15, 6-17, 6-25 (col 14, lines 50-67 and col 15, lines 1-48) and 6-26, together performs the function of the NSP)** to a second Software Radio Port (SRP) wherein the mobile station is connected with a party, the method comprising:

sending a handoff request from the first SRP to a Network Server Platform (NSP) (Fig. 14, Col 23, lines 43-50, Handover Req, LRQ[BTS_ID,CPN,32Kkps,CRV] in Phase 1

discusses sending a handoff request from the first SRP to a Network Server Platform (NSP));

handing off the mobile station from the first SRP to the second SRP via the NSP (Fig. 14, Col 23, lines 43-67, Col 24, lines 1-17. Phase 1 through Phase 5 discuss handing off the mobile station from the first SRP to the second SRP via the NSP);

sending a call transfer request from the first SRP to the NSP (Fig. 14, Col 23, lines 51-67. NonStandardMessage[Ho_Request,H.323_ID=HO Info] in Phase 2 discusses sending a call transfer request from the first SRP to the NSP);

releasing radio resources at the first SRP (Fig. 14, Col 24, lines 11-17 End Session in Phase 5 discusses releasing radio resources at the first SRP) ;

detecting at the second SRP the mobile station as being tuned to a digital traffic channel and sending a conference call request to the party via a VoIP call-server (Fig. 14, Col 24, lines 1-10. In Phase 3 and Phase 4 the NonStandardMessage[Ho_Accept,H.323_ID=HO Info], Setup, Connect, Release Complete and Bridged messages are related to detecting at the second SRP the mobile station as being tuned to a digital traffic channel and sending a conference call request to the party via a VoIP call-server);

setting up an RTP media path for exchange of RTP data packets via the second SRP when the conference call has been established (Fig. 14, Col 24, lines 5-10. Phase 4 indicates once the multicast distributed conference has been established the old BTS can instruct the mobile station to handover to the target BTS. Fig. 14 shows after the Bridged message an RTP/RTCP Channel open between endpoints);

interconnecting the voice path between the second SRP and the mobile station and the RTP path (Fig. 14, Col 24, lines 11-17. In Phase 5, the Handover Command is sent to the mobile station which will begin to access the Target BTS on the indicated timeslot. The RTP/RTCP Channel open between Endpoints from MS to EP2 H.323 (shown at the top of Fig. 14) and RTP/RTCP Channel open between Endpoints from EP2 H.323 to EP3-New-P-BTS H.323 (shown below in Fig. 14) is associated with interconnecting the voice path between the second SRP and the mobile station and the RTP path);

sending a handoff complete indication from the second SRP to the NSP (Fig. 14, Col 24, lines 5-10. The Connect(EP3,H.245.TSAP) and Release Complete in Phase 4 is associated with sending a handoff complete indication from the second SRP to the NSP);

sending a call release request from the first SRP to the party via the VoIP call-server (Fig. 14, Col 24, lines 11-17. In Phase 5, the Access message shown in Fig. 14 is associated with sending a call release request from the first SRP to the party via the VoIP call-server);

releasing the RTP media path at the first SRP; and sending call release indication from the first SRP to the NSP (Fig. 14, Col 24, lines 11-17. In Phase 5, the Physical Info, Handover Complete, Handover Com and End Session shown in Fig. 14 are associated with releasing the RTP media path at the first SRP; and sending call release indication from the first SRP to the NSP).

Claim Rejections - 35 USC § 103

6. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

7. Claims 15-24 (previously claims 14-22), 34-44 (previously claims 32-42) are rejected under 35 U.S.C. 103(a) as being unpatentable over Sayers et al. (U.S. Patent No. 6,539,237) in view of Christie, IV (U.S. Patent No. 6,430,176).

Regarding Claim 15 (previously Claim 14), Sayers et al. discloses a method of providing a two-way voice path between a VoIP device (Fig. 4, gateway 42) in a network and a mobile station (Fig. 4 shows mobile station (4), col 11, lines 43-47) wherein a call originates at the mobile station, the method comprising: initiating mobile call set-up (Fig. 10 and Fig. 11, col 19, lines 65-67 and col 20, lines 1-7. Phase 1 and Phase 2 discuss the Connection Management (CM) Service request and SETUP message); tuning the mobile station to digital traffic channel (DTC) to establish a voice path over the air (Fig. 10, col 20, lines 45-50. Phase 6 relates to the assignment of the mobile station to the traffic channel (TCH). The mobile station operates in a wireless network and is therefore considered to be communication over the air. This is associated with tuning the mobile station to digital traffic channel to establish a voice path over the air); engaging a VoIP call-server to set up a VoIP call (Fig. 10 and Fig. 11,

col 20, lines 8-36. Phase 3, Phase 4 discuss the CM layer begin access process in the H.323 stack towards the Gatekeeper. H.323 relates to VoIP protocol, therefore the Gatekeeper is the VoIP call-server); establishing an RTP media path for exchange of RTP data packets (Fig. 10, col 20 lines 37-44. Phase 5 discusses B-Open Chn with RTP TSAP which provides RTP TSAP address to be used for actual data transfer); and interconnecting the voice path over the air and the RTP path over the packet network (Fig. 11 shows RTP/RCTP Channel open between Endpoints for communication).

Sayers et al. discloses all the limitations of the claims except for generating a ringback tone to the mobile station. Christie, IV discloses the sequence of messages sent between User1 (element 20, calling party) and User2 (element 32, H.323 terminal device, called party) during the establishment of the multimedia communication sessions (Fig. 3). A Ring Back message (element 125) is sent back to the calling party from the called party upon receiving alerting message (element 124) at the computer telephony integration (CTI) server (element 42). See col 6, lines 42-56. It would have been obvious to one of ordinary skill in the art at the time the invention was made to incorporate the Christie, IV's teaching into the Sayers et al. to provide ring notification message while said multimedia communication session is being established.

Regarding Claim 16 (previously Claim 14.5), Sayers et al. discloses wherein the VoIP device comprises at least one of a VoIP phone or a VoIP Gateway (Fig. 4, gateway 42, col 11, lines 37-40).

Regarding Claim 17 (previously Claim 15), Sayers et al. discloses wherein the step of initiating mobile call set-up comprises receiving a call origination message from the mobile station and engaging the NSP to set up a mobile call (Fig. 6 elements 6-1 through 6-15, 6-17, 6-25 (col 14, lines 50-67 and col 15, lines 1-48) and 6-26, together performs the function of the NSP. Fig. 10 and Fig. 11 refers to the call origination process as discuss in Phase 1 through Phase 6 in col 19, lines 65-67 and col 20 lines 1-50. The MM, CM, H.225.0, RAS, H.425 and Gatekeeper are part of the elements referred to in Fig. 6 and Fig. 7 that forms the NSP).

Regarding Claim 18 (previously Claim 16), Sayers et al. discloses wherein said step of tuning comprises: sending a message to tune the mobile station to a specified digital traffic channel (Fig. 10 (see Asg CMD and Asg COM), col 20, lines 45-50. Phase 6 relates to the assignment of the mobile station to the traffic channel (TCH). This is associated with tuning the mobile station to digital traffic channel); and detecting the mobile station as being tuned to the specified digital traffic channel (In Fig. 11 the messages H.323-Alerting [H.245.TSAP, CRV] and H.323-Connect [CRV] from EP to H.225.0, ALERTING and CONNECT from CM to MS and CONNECT ACK from MS to CM are the processes involved in detecting the mobile station as being tuned to the specified digital traffic channel).

Regarding Claim 19 (previously Claim 17), Sayers et al. discloses wherein said step of engaging comprises: sending a VoIP call connection request to the VoIP call-server (Fig. 10, col 20, lines 8-10. The CM layer will begin the access process in the H.323 stack by using the RAS Admission Request (ARQ) sequence towards the P-BTS/gatekeeper); analyzing a called number (col 20, lines 17-21. The ARQ generated by the P-BTS will contain the Called Party Number (CPN). The CPN number is used by the gatekeeper to determine the process to be followed in establishing the call. This process is related to analyzing a called number); and setting up a VoIP call via the VoIP call-server (Fig. 10, col 20, lines 30-44).

Regarding Claim 20 (previously Claim 18), Sayers et al. fails to disclose wherein said step of generating comprises: receiving a ringing indication from the called party; generating a ringback tone in response to said receiving; and transmitting the ringback tone to the mobile station. Christie, IV discloses the sequence of messages sent between User1 (element 20, calling party) and User2 (element 32, H.323 terminal device, called party) during the establishment of the multimedia communication sessions (Fig. 3). The H.323 terminal device 32 sends alerting message 116 (first alerting messages 116a and 116b) to User2 CO 26, through Gateway 44. At CO 26, the relevant information contained in alerting message 116 is inserted into another ISUP message called the address complete message (ACM) 122 which is transmitted across PSTN 14 (not shown) to CO 24. CO 24 then transmits a second alerting message 124 to CTI server 42 (this is associated with generating a ringback tone in response to said

receiving). CTI server 42 transmits a ring back message 125 to User1 telephone 20 (this is associated with transmitting the ringback tone to the mobile station). User1 may have selected a service that provides User1 with a ringing sound to alert User1 that User2 H.323 terminal device 32 is ringing (this is associated with receiving a ringing indication from the called party). See col 6, lines 42-56. It would have been obvious to one of ordinary skill in the art at the time the invention was made to incorporate the Christie, IV's teaching into the Sayers et al. to provide ring notification message while said multimedia communication session is being established.

Regarding Claim 21 (previously Claim 19), Sayers et al. discloses wherein said step of establishing comprises: receiving a connect indication from the called party (In Fig. 11 the messages H.323-Alerting [H.245.TSAP, CRV] and H.323-Connect [CRV] from EP to H.225.0, ALERTING and CONNECT from CM to MS are the processes that indicates receiving a connect indication from the called party); turning off the ringback tone (CONNECT ACK from MS to CM acknowledgement turns off the ringback tone); setting up an RTP media path for exchange of RTP data packets (Fig. 10, col 20, lines 37-44. Phase 5 discusses B-Open Chn with RTP TSAP which provides RTP TSAP address to be used for actual data transfer); and informing the NSP of the call connection (In Fig. 11, the CONNECT ACK to the CM indicates the call connection where the CM is part of the NSP).

Regarding Claim 22 (previously Claim 20), Sayers et al. discloses wherein said step of interconnecting comprises: converting received voice frames to RTP packets to be sent to the packet network, and converting received RTP packets to voice frames to be sent to the mobile station (Fig. 3 - Col 10, lines 56-59, Fig. 5 - Col 11, lines 55-60 and Fig. 10 – Col 20, lines 37-44. Air Interface (31-1) /GSM Um-Interface (31-1') and Network Interface (31-2)/H.323 Terminal Interface (31-2') converts voice frames into packets and converts received packets to voice frames for transmission in the RTP/RTCP transport link between the endpoints as shown in Fig. 11 (MS and EP)).

Regarding Claim 23 (previously Claim 21), Sayers et al. discloses wherein the network is a Public Switched Telephone Network (PSTN) (Fig. 4, element 26 (PSTN), col 11, lines 37-43) .

Regarding Claim 24 (previously Claim 22), Sayers et al. discloses wherein the network is a Private Branch Exchange (PBX) (Fig. 4, element 43 (PBX), col 11, lines 37-43).

Regarding Claim 34 (previously Claim 32), Sayers et al. discloses a method of providing a two-way voice path between a first mobile station and a second mobile station **(Note: first mobile station (origination) to second mobile station (termination) can be explained using the combination of Fig. 10 and Fig. 11 (relates to the first mobile station) and Fig. 12 and Fig 13 (relates to the second**

mobile station)) wherein the first mobile station is associated with a first Software Radio Port (SRP) **(Note: SRP consists of the NSP elements and includes the air interface modules as defined in Fig. 3 and Fig. 5 for wireless communication with the mobile station. Note: Fig. 6 elements 6-1 through 6-15, 6-17, 6-25 (col 14, lines 50-67 and col 15, lines 1-48) and 6-26, together performs the function of the NSP)** and the second mobile station is associated with a second SRP and wherein a call originates at the first mobile station, the method comprising:

initiating call set-up for the first mobile station at the first SRP (Fig. 10 and Fig. 11, col 19, lines 65-67 and col 20, lines 1-7. Phase 1 and Phase 2 discuss the Connection Management (CM) Service request and SETUP message for the first mobile station at the first SRP);

tuning the first mobile station to a digital traffic channel (DTC) via the first SRP to establish a voice path over the air (Fig. 10, col 20, lines 45-50. Phase 6 relates to the assignment of the first mobile station to the traffic channel (TCH). This is associated with tuning the first mobile station to digital traffic channel to establish a voice path over the air via the first SRP);

engaging a VoIP call-server to set up a VoIP call via the first SRP (Fig. 10 and Fig. 11, col 20, lines 8-36. Phase 3, Phase 4 discuss the CM layer begin access process in the H.323 stack towards the Gatekeeper. H.323 relates to VoIP protocol, therefore the Gatekeeper is the VoIP call-server via the first SRP);

initiating mobile call set-up for the second mobile station via a Network Server Platform (NSP) (Fig. 12, col 22, lines 4-21. Phase 2 and Phase 3 discuss the process of

initiating a mobile call set-up at a NSP starting with the Paging Request and ending when the Call Confirmed message is received from the second mobile station) ;

tuning the second mobile station to a digital traffic channel (DTC) via the second SRP to establish a voice path over the air (Fig. 12, col 22, lines 22-27. Phase 4 discusses the Terminal Capabilities, Master/Slave processes and assignment of the second mobile station to the traffic channel (TCH) shown in Fig. 12 by Asg CMD and Asg COM (refer to Fig. 10 Asg CMD and Asg COM). This is associated with tuning the second mobile station to digital traffic channel to establish a voice path over the air via the second SRP);

alerting the first mobile station and the second mobile station via the second SRP (In Fig. 11 H.323-Alerting [H.245.TSAP, CRV] , H.323-Connect [CRV] and ALERTING messages correspond to alerting the first mobile station and in Fig 12 the Asg CMD can be associated with Alerting the second mobile station via the second SRP);

establishing an RTP media path for exchange of RTP data packets (Fig. 10, col 20 lines 37-44, Fig. 12, col 22, lines 22-27. Phase 5 in Fig. 10 and Phase 4 in Fig. 12 discuss B-Open Chn with RTP TSAP which provides RTP TSAP address to be used for actual data transfer);

interconnecting a voice path between the first SRP and the first mobile station and an RTP path over the packet network (Fig. 11 shows RTP/RCTP Channel open between Endpoints for communication);

and interconnecting a voice path between the second SRP and second mobile station and an RTP path over the packet network (Fig. 13 shows RTP/RCTP Channel open between Endpoints for communication).

Sayers et al. discloses all the limitations of the claims except for generating a ringback tone to the first mobile station via the first SRP. Christie, IV discloses the sequence of messages sent between User1 (element 20, calling party) and User2 (element 32, H.323 terminal device, called party) during the establishment of the multimedia communication sessions (Fig. 3). A Ring Back message (element 125) is generated at the CTI (element 42) and is transmitted to the calling party. See col 6, lines 42-56. It would have been obvious to one of ordinary skill in the art at the time the invention was made to incorporate the Christie, IV's teaching into the Sayers et al.'s SRP wherein the function of the SRP through the use of the Call Control Module and Tone and Announcement Module provide ring notification message to the mobile station while said multimedia communication session is being established.

Regarding Claim 35 (previously Claim 33), Sayers et al. discloses wherein the step of initiating call set-up for the first mobile station at the first SRP comprises receiving a call origination message from the first mobile station and engaging the NSP to set up a mobile call (Fig. 10 and Fig. 11 refers to the call origination process as discuss in Phase 1 through Phase 6 in col 19, lines 65-67 and col 20 lines 1-50 from the first mobile station at the first SRP and engaging the NSP to set up a mobile call).

Regarding Claim 36 (previously Claim 34), Sayers et al. discloses wherein said step of tuning the first mobile station to a digital traffic channel (DTC) via the first SRP comprises: sending a message to tune the first mobile station to a specified digital traffic channel (Fig. 10 (see Asg CMD and Asg COM), col 20, lines 45-50. Phase 6 relates to the assignment of the first mobile station to the traffic channel (TCH). This is associated with tuning the first mobile station to digital traffic channel via the first SRP); and detecting the first mobile station as being tuned to the digital traffic channel (In Fig. 11 the messages H.323-Alerting [H.245.TSAP, CRV] and H.323-Connect [CRV] from EP to H.225.0, ALERTING and CONNECT from CM to MS and CONNECT ACK from MS to CM are the processes involved in detecting the first mobile station as being tuned to the specified digital traffic channel).

Regarding Claim 37 (previously Claim 35), Sayers et al. discloses wherein said step of engaging a VoIP call-server to set up a VoIP call comprises:
sending a VoIP call connection request to the VoIP call-server (Fig. 10, col 20, lines 8-10. The CM layer will begin the access process in the H.323 stack by using the RAS Admission Request (ARQ) sequence towards VoIP call-server); and
coordinating with the NSP to analyze a called number via the VoIP call server (col 20, lines 17-21. The ARQ generated by the P-BTS will contain the Called Party Number (CPN). The CPN number is used by the gatekeeper (VoIP call server) to determine the

process to be followed in establishing the call. This process is coordinated with the NSP and is associated with analyzing a called number).

Regarding Claim 38 (previously Claim 36), Sayers et al. discloses wherein said step of initiating mobile call set-up for the second mobile station via a Network Server Platform comprises: verifying the called number as a valid mobile station (Fig. 12, Col 22, lines 4-11. Phase 2 discusses authentication procedure to verify the called party as a valid mobile station); sending a message to page the second mobile station via the second SRP (Fig. 12, Col 22, lines 4-11. In Phase 2, a Paging Request is sent to the second MS via the second SRP); receiving a page response from the second mobile station (Fig. 12, Col 22, lines 4-11. As part of Phase 2, a paging response from the second MS is associated with a paging request (see message Paging Response[IMSI,Cksn] in Fig. 12)); and instructing the VoIP call-server to forward the call connection request to the second SRP (Fig. 12, Col 22, lines 4-11. In Phase 2, the ARQ and ACF instruct the VoIP call-server to forward the call connection request to the second SRP. Note: H.323 relates to VoIP protocol, therefore the Gatekeeper is the VoIP call-server).

Regarding Claim 39 (previously Claim 37), Sayers et al. discloses wherein said step of tuning the second mobile station to a digital traffic channel (DTC) via the second SRP comprises: sending a message to tune the second mobile station to a specified digital traffic channel (Fig. 12, col 22, lines 22-27. Phase 4 discusses the Terminal

Capabilities, Master/Slave processes and assignment of the mobile station to the traffic channel (TCH) shown in Fig. 12 by Asg CMD and Asg COM (refer to Fig. 10 Asg CMD and Asg COM). This is associated with tuning the second mobile station to digital traffic channel); and detecting the second mobile station as being tuned to the specified digital traffic channel (Fig. 12, col 22, lines 28-33. In Phase 5, the ALERTING, H.323-Alerting, CONNECT, H.323-Connect and CONNECT ACK messages are the processes involved in detecting the second mobile station as being tuned to the specified digital traffic channel).

Regarding Claim 40 (previously Claim 38), Sayers et al. discloses wherein said step of alerting the first mobile station and the second mobile station comprises:
sending a message to the second mobile station for alerting a user (In Fig. 12 the Asg CMD can be associated with Alerting the MS); and
sending a ringing indication to the first SRP via the VoIP call-server (Fig. 12 col. 22, lines 28-33. In Fig 12, ALERTING and H.323-Alerting and Fig 11, H.323-Alerting [H.245.TSAP, CRV] and ALERTING messages are associated with ringing indication to the first SRP via the VoIP call-server).

Regarding Claim 41 (previously Claim 39), Sayers et al. fails to disclose wherein said step of generating a ringback tone to the first mobile station via the first SRP comprises: receiving a ringing indication from the called party; generating a ringback tone in response to said receiving; and transmitting the ringback tone to the first mobile

station. Christie, IV discloses the sequence of messages sent between User1 (element 20, calling party) and User2 (element 32, H.323 terminal device, called party) during the establishment of the multimedia communication sessions (Fig. 3). The H.323 terminal device 32 sends alerting message 116 (first alerting messages 116a and 116b) to User2 CO 26, through Gateway 44. At CO 26, the relevant information contained in alerting message 116 is inserted into another ISUP message called the address complete message (ACM) 122 which is transmitted across PSTN 14 (not shown) to CO 24. CO 24 then transmits a second alerting message 124 to CTI server 42 (this is associated with generating a ringback tone in response to said receiving). CTI server 42 transmits a ring back message 125 to User1 telephone 20 (this is associated with transmitting the ringback tone to the mobile station). User1 may have selected a service that provides User1 with a ringing sound to alert User1 that User2 H.323 terminal device 32 is ringing (this is associated with receiving a ringing indication from the called party). See col 6, lines 42-56. It would have been obvious to one of ordinary skill in the art at the time the invention was made to incorporate the Christie, IV's teaching into the Sayers et al.'s SRP wherein the function of the SRP through the use of the Call Control Module and Tone and Announcement Module provide ring notification message to the mobile station while said multimedia communication session is being established.

Regarding Claim 42 (previously Claim 40), Sayers et al. discloses wherein said step of establishing an RTP media path for exchange of RTP data packets comprises:

receiving a connect indication at the second SRP from the second mobile station; sending a connect indication from the second SRP to the VoIP call-server (Fig. 12 col 22, lines 28-33. The CONNECT and H.323-Connect show the connect indication received at the second SRP from the second mobile station and sending connect indication from the second SRP to the the VoIP call-server (gatekeeper)); receiving a connect indication at the first SRP from the VoIP call-server (In Fig. 11 the messages H.323-Alerting [H.245.TSAP, CRV] and H.323-Connect [CRV] , ALERTING and CONNECT are the processes that indicates receiving a connect at the first SRP from the VoIP call-server); sending back an acknowledge message from the first SRP (In Fig. 11, the CONNECT ACK message is related to sending back an acknowledge message from the first SRP) ; turning off the ringback tone (CONNECT ACK message is associated with turning off the ringback tone); setting up the RTP media path for exchange of RTP data packets (Fig. 10, col 20, lines 37-44, Fig. 12, col 22, lines 22-27. Phase 5 (Fig. 10) and Phase 4 (Fig. 12) discuss the B-Open Chn with RTP TSAP which provides RTP TSAP address to be used for actual data transfer); and informing the NSP of the call connection (In Fig. 12, the CONNECT to the CM indicates the call connection where the CM is part of the NSP. In Fig. 11, the CONNECT ACK to the CM indicates the call connection where the CM is part of the NSP).

Regarding Claim 43 (previously Claim 41), Sayers et al. discloses wherein said step of interconnecting a voice path between the first SRP and the first mobile station

and an RTP path over the packet network comprises: converting received voice frames from the first mobile station to RTP packets to be sent to the packet network; and converting received RTP packets to voice frames to be sent to the first mobile station (Fig. 3 - Col 10, lines 56-59, Fig. 5 - Col 11, lines 55-60 and Fig. 10 – Col 20, lines 37-44. Air Interface (31-1) /GSM Um-Interface (31-1') and Network Interface (31-2)/H.323 Terminal Interface (31-2') converts voice frames into packets and converts received packets to voice frames for transmission in the RTP/RTCP transport link between the endpoints as shown in Fig. 11).

Regarding Claim 44 (previously Claim 42), Sayers et al. discloses wherein said step of interconnecting a voice path between the second SRP and second mobile station and an RTP path over the packet network comprises: converting received voice frames from the second mobile station to RTP packets to be sent to the packet network; and converting received RTP packets to voice frames to be sent to the second mobile station (Fig. 3 - Col 10, lines 56-59, Fig. 5 - Col 11, lines 55-60 and Fig. 10 and Fig. 12 – Col 22, lines 22-27. Air Interface (31-1) /GSM Um-Interface (31-1') and Network Interface (31-2)/H.323 Terminal Interface (31-2') converts voice frames into packets and converts received packets to voice frames for transmission in the RTP/RTCP transport link between the endpoints as shown in Fig. 13).

Conclusion

8. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

U.S. Patent No. 6,885,658 to Ress et al. discusses a method and apparatus for interworking between internet protocol telephony protocols relating to MGCP and H.323.

U.S. Patent No. 6, 888,803 to Gentry et al. relates to a system for providing connectivity of wireless base station to PSTN via an IP data network.

U.S. Patent No. 6,404,746 to Cave et al. relates to a system and method for a packet voice response unit utilizing packet network protocols, such as H.323. for packet network media redirection.

U.S Patent No. 6,434,140 to Barany et al. relates to a system and method for implementing XOIP over ANSI-136-A circuit/switched/packet-switched mobile communications networks.

The above prior art are cited to further show the state of the art with respect to wireless mobile communication using IP, H.323, RTP protocols.

9. Any inquiry concerning this communication or earlier communications from the examiner should be directed to F. Lin Khoo whose telephone number is 571-272-5508. The examiner can normally be reached on flex time.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Wellington Chin can be reached on 571-272-3134. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

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